EXAMINER'S AMENDMENT

Continued Examination Under 37 CFR 1.114

1. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on 14 April 2009 has been entered.

2. An examiner's amendment to the record appears below. Should the changes and/or additions be unacceptable to applicant, an amendment may be filed as provided by 37 CFR 1.312. To ensure consideration of such an amendment, it MUST be submitted no later than the payment of the issue fee.

Authorization for this examiner's amendment was given in a telephone interview with Martin Miller on 25 June 2009.

The application has been amended as follows:

In the Specification, first paragraph:

This application is a continuation of application serial number 09/628,980, filed February 11, 2000 now U.S. Patent No. 6,775,270. Application serial number 09/628,980 is a continuation of application serial number 09/326,265, filed June 7, 1999 and now abandoned.

In the claims:

46. (Currently Amended) A method of setting up a voice call between two digital wireless telephones, comprising:

receiving, at a switch in a communication network, a call setup request from a calling digital wireless telephone<u>, wherein the communication network is a circuit switched network</u>;

determining, prior to completing the call setup between the calling digital wireless telephone and a called digital wireless telephone, whether the called wireless digital telephone uses a voice compression algorithm that is compatible with a voice compression algorithm used by the calling digital wireless telephone;

if the called digital wireless telephone and the calling digital wireless telephone use incompatible voice compression algorithms, setting up the voice call over the communication network;

if the called digital wireless telephone and the calling digital wireless telephone use compatible voice compression algorithms, (a) setting up the voice call over the Internet; and

(b) exchanging compressed voice signals between the calling digital wireless telephone and the called digital wireless telephone during the voice call over the Internet, wherein the voice signals are compressed using the compatible voice compression algorithm at either the calling digital wireless telephone or the called digital wireless telephone.

47. (Canceled).

53. (Currently Amended) A method comprising:

initiating a call setup between a calling party's audio device and a called party's audio device using a first path, wherein the first path includes a public switched telephone network;

before the call setup is completed, determining whether the called party's audio device supports a voice compression algorithm compatible with a voice compression algorithm supported by the calling party's audio device;

if the calling party's audio device and the called party's audio use compatible voice compression algorithms, completing the call setup using a second path different from the first path, wherein the second path includes a data network; and

if the calling party's audio device and the called party's audio device use incompatible voice compression algorithms, maintaining the call from the calling party's audio device to the called party's audio device on the first path.

54-55. (Canceled).

61. (Currently Amended) A method comprising:

sending a call request message to a called digital wireless telephone via a public switched telephone network as a portion of a call setup procedure, the call request message including a list of voice compression algorithms supported by a calling digital wireless telephone;

receiving a response message indicating whether the called digital wireless telephone supports one of the voice compression algorithms on the list, and whether the called digital wireless telephone accesses a data network also accessible to the calling digital wireless telephone;

if the response message indicates that the called digital wireless telephone ~~can support~~supports one of the voice compression algorithms on the list and that the called digital wireless telephone ~~can access~~accesses the data network, completing the call setup procedure via the data network and sending data compressed according to the listed voice compression algorithm from the calling digital wireless telephone to the called digital wireless telephone; and

if the response message indicates that the called digital wireless telephone ~~cannot~~does not support one of the voice compression algorithms on the list or that the called digital wireless telephone ~~cannot~~does not access the data network, completing the call setup procedure via the public switched telephone network.

65. (Currently Amended) A method for diverting an Integrated Services Digital Network User Part (ISUP) network talkpath to a data network talkpath, comprising the steps of:

determining, using an ISUP signaling path, whether a called party's telephone is configured to exchange voice signals via a same data network to which a calling party's telephone is adapted to exchange voice signals using a voice compression algorithm compatible with both the calling party's telephone and the called party's telephone, said ISUP signaling path being established during a process of establishing the ISUP network talkpath;

establishing the data network talkpath using resources associated with said same data network if said called party's telephone is configured to exchange compressed voice signals

between said calling party's telephone and said called party's telephone via said same data network using the compatible voice compression algorithm; ~~and~~

exchanging voice signals between said called party's telephone and said calling party's telephone using the data network talkpath; and

otherwise, using the ISUP network talkpath if neither the called party's telephone and the calling party's telephone are adapted to exchange voice signals using a voice compression algorithm compatible with both the calling party's telephone and the called party's telephone.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to CHRISTINE DUONG whose telephone number is (571)270-1664. The examiner can normally be reached on Monday - Friday: 830 AM-6 PM EST with first Friday off.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Seema Rao can be reached on (571) 272-3174. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/Seema S. Rao/
Supervisory Patent Examiner, Art Unit 2416

/Christine Duong/
Examiner, Art Unit 2416
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